

SUPPLEMENT

TO BRITISH TELECOMMUNICATIONS ENGINEERING

(formerly the Supplement to The Post Office Electrical Engineers' Journal)

Vol. 5 Part 2 July 1986

ISSN 0262-4028

BTEC & CGLI
GUIDANCE FOR STUDENTS

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BUSINESS AND TECHNICIAN EDUCATION COUNCIL

National Certificate in Telecommunications

Sets of model questions and answers for Business and Technician Education Council (BTEC) units are given below. The questions illustrate the types of questions that students may encounter, and are useful as practice material for the skills learned during the course.

Where additional text is given for educational purposes, it is shown within square brackets to distinguish it from information expected of students under examination conditions. Representative time limits for questions are shown, and care has been taken to give model answers that reflect these limits.

We would like to emphasise that the questions are not representative of questions set by any particular college.

BTEC: DIGITAL TECHNIQUES II

The questions in this paper are based on the BTEC's standard unit U8/750, Students are advised to read the notes above

Q1 What is the difference in value between the decimal number 0.1 and the binary number 0.1? Express your answer in decimal. (4 min)

A1 Since the answer is to be expressed in decimal, the binary number is converted to decimal as follows:

$$\begin{array}{c} 0.1 \\ \downarrow \\ (1 \times 2^{-1}) = 0.5 \end{array}$$

Thus, the difference is $0.5 - 0.1 = 0.4$.

Q2 What is meant by the 'weight' of a binary digit? Illustrate your answer by converting the binary number 1111 to decimal. (3 min)

A2 The 'weight' of a binary digit is the decimal value attributed to the digit because of its position in a binary number. These 'weights' start with a value of 1 given to the least significant bit of a binary integer and double for each bit position in the number moving towards the most significant bit; that is, the 'weight' represents increasing powers of 2 for each bit position in the number.

This can be illustrated as follows:

$$\begin{array}{cccc} 1 & 1 & 1 & 1 \\ \downarrow & \downarrow & \downarrow & \downarrow \\ (1 \times 2^3) & + (1 \times 2^2) & + (1 \times 2^1) & + (1 \times 2^0) = 15 \end{array}$$

The least significant bit represents multiples of 2^0 , the next digit represents multiples of 2^1 , and so on. The decimal equivalent is obtained by adding the weighted digits together.

Q3 (a) Briefly describe how negative numbers can be represented in computers.

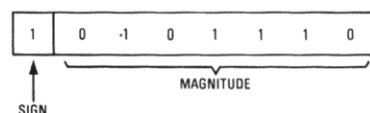
(b) Convert the following numbers to binary:

- (i) -125, and
(ii) -63.

(7 min)

A3 (a) Negative numbers are generally represented by using two's complement notation. In this form of representation, the sign of a number is indicated by the most significant bit, with a logic 1 representing a negative number. The rest of the number represents its magnitude. See sketch.

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(b) The first stage in the conversion is to represent each positive decimal number in binary. This can be done by repeatedly dividing each one by 2 and noting the remainder at each stage.

$$\begin{array}{r} \text{(i)} \quad 2) \quad 125 \\ \quad \quad 62 \quad 1 \\ \quad \quad 31 \quad 0 \\ \quad \quad 15 \quad 1 \\ \quad \quad 7 \quad 1 \\ \quad \quad 3 \quad 1 \\ \quad \quad 1 \quad 1 \\ \quad \quad 0 \quad 1 \end{array}$$

Thus, 125 in 8 bit binary is 01 111 101, with a sign bit added.

$$\begin{array}{r} \text{(ii)} \quad 2) \quad 63 \\ \quad \quad 31 \quad 1 \\ \quad \quad 15 \quad 1 \\ \quad \quad 7 \quad 1 \\ \quad \quad 3 \quad 1 \\ \quad \quad 1 \quad 1 \\ \quad \quad 0 \quad 1 \end{array}$$

Thus, 63 in 8 bit binary is 00 111 111, with a sign bit added.

The representation of the negative numbers is achieved by taking the two's complement of each number. This involves inverting each binary number and adding 1.

$$\begin{array}{r} \text{(i)} \quad \text{Number: } 0111101 \\ \quad \quad \text{Invert: } 1000010 \\ \quad \quad \text{Add 1: } 1000011 \end{array}$$

Thus -125 is represented by 1000011 in 8 bit binary.

$$\begin{array}{r} \text{(ii)} \quad \text{Number: } 0011111 \\ \quad \quad \text{Invert: } 1100000 \\ \quad \quad \text{Add 1: } 1100001 \end{array}$$

Thus -63 is represented by 1100001 in 8 bit binary.

Q4 Which of the answers given below represents

- (a) the sum, and
(b) the difference

of the binary numbers 1001 100 011-101 and 110 101 110-011?

- (i) 10 010 010 010-000
(ii) 00 010 110 101-010
(iii) 00 101 101 010-100
(iv) 10 000 010 010-000
(v) 10 010 010 000-010
(vi) 00 010 110 101-000

A4 (a) The sum is found as follows

$$\begin{array}{r} 1001\ 100\ 011-101 \\ 110\ 101\ 110-011 \\ \hline \text{Carry: } 11\ 111\ 11\ 111\ 11 \\ \text{Sum: } 10000\ 010\ 010-000 \end{array}$$

Thus answer (iv) is correct.

(b) The difference is found as follows:

$$\begin{array}{r} 1001\ 100\ 011-101 \\ 110\ 101\ 110-011 \\ \hline \text{Borrow: } 11\ 1111\ 1 \\ \text{Difference: } 0010\ 110\ 101-010 \end{array}$$

Thus answer (ii) is correct, although it has a leading 0 added.

[Tutorial Note: The difference could have been obtained by using an alternative method such as complement addition.]

Q5 Use a method of complement addition to subtract the following unsigned binary numbers:

111 000 010 and 10 111 100 (4 min)

A5 First, the complement of the second number is found. The second number must have the same number of bits as the first number; that is by adding a leading 0:

$$\begin{array}{r} \text{Invert: } 010\ 111\ 100 \\ \text{Add 1: } 101\ 000\ 011 \\ \hline \text{Complement: } 101\ 000\ 100 \\ \text{Add: } 111\ 000\ 010 \\ \hline \text{Difference: } 1\ 100\ 000\ 110 \end{array}$$

Ignore the overflow bit to give the result 100 000 110.

Q6 Perform the calculation 27×13 in binary. (5 min)

A6 First each number is converted to binary.

Thus, 27 becomes 11 011, and
13 becomes 1 101

Multiply them together:

$$\begin{array}{r} 11\ 011 \\ 1\ 101 \\ \hline 11\ 011 \quad \text{First partial product} \\ 1\ 101\ 100 \quad \text{Second partial product} \\ \hline 10\ 000\ 111 \quad \text{Add} \\ 11\ 011\ 000 \quad \text{Third partial product} \\ \hline 101\ 011\ 111 \quad \text{Add Final product} \end{array}$$

Thus, the result is 101 011 111 in binary, or 351 in decimal.

Q7 Which of the following binary numbers divide into 10 100 101 without any remainder?

- (a) 1111
(b) 1101
(c) 1011

A7 (a)

$$\begin{array}{r} 1011 \\ 1111 \overline{)10\ 100\ 101} \\ \underline{1111} \\ 101\ 10 \\ \underline{1111} \\ 1111 \\ \underline{1111} \\ 0000 \end{array}$$

The remainder is 0. Thus, answer (a) is correct.

(b)

$$\begin{array}{r} 1100 \\ 1101 \overline{)10\ 100\ 101} \\ \underline{1101} \\ 111\ 1 \\ \underline{1101} \\ 1001 \end{array}$$

The remainder is 1001. Thus, answer (b) is not correct.

(8 min)

(c)

$$\begin{array}{r} 1111 \\ 1011 \overline{)10\ 100\ 101} \\ \underline{1011} \\ 1001\ 1 \\ \underline{1011} \\ 100\ 00 \\ \underline{1011} \\ 1011 \\ \underline{1011} \\ 0000 \end{array}$$

The remainder is 0. Thus, answer (c) is correct.

Q8 The output of a parity-checking circuit is logic 1 if there is an odd number of logic 1s at its four inputs. Draw the truth table for the circuit. (4 min)

A8 The truth table is given below:

| Inputs | | | | Output |
|--------|---|---|---|--------|
| A | B | C | D | |
| 0 | 0 | 0 | 0 | 0 |
| 0 | 0 | 0 | 1 | 1 |
| 0 | 0 | 1 | 0 | 1 |
| 0 | 0 | 1 | 1 | 0 |
| 0 | 1 | 0 | 0 | 1 |
| 0 | 1 | 0 | 1 | 0 |
| 0 | 1 | 1 | 0 | 0 |
| 0 | 1 | 1 | 1 | 1 |
| 1 | 0 | 0 | 0 | 1 |
| 1 | 0 | 0 | 1 | 0 |
| 1 | 0 | 1 | 0 | 0 |
| 1 | 0 | 1 | 1 | 1 |
| 1 | 1 | 0 | 0 | 0 |
| 1 | 1 | 0 | 1 | 1 |
| 1 | 1 | 1 | 0 | 1 |
| 1 | 1 | 1 | 1 | 0 |

Q9 The logic circuit shown in Fig. 1 has been constructed by using electromechanical relays for the switching elements. It must be replaced by an electronic logic circuit to perform the same function. Draw a suitable circuit showing the gates required and write down the Boolean expression for the circuit. (6 min)

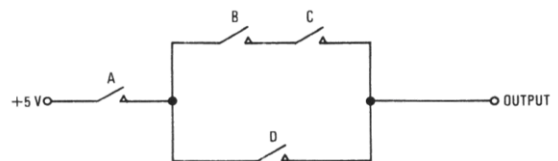


Fig. 1

A9 The Boolean expression for the circuit is

$$A \cdot ((B \cdot C) + D),$$

which can be represented by the logic circuit shown in the sketch.

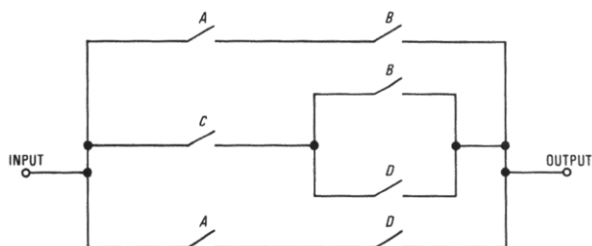


[Tutorial Note: Other circuits can be devised which would perform the same logical function with different gates.]

Q10 Draw a simple switch circuit that performs the following logical operation:

$$(A \cdot B) + C \cdot (B + D) + (A \cdot D) \quad (4 \text{ min})$$

A10 The required switch circuit has three branches, one for each of the terms of the above expression. Thus the circuit is as shown in the sketch.



Q11 Derive the truth table for the logic circuit given in Q10. (4 min)

A11 The truth table is given below

| A | B | C | D | Output |
|---|---|---|---|--------|
| 0 | 0 | 0 | 0 | 0 |
| 0 | 0 | 0 | 1 | 0 |
| 0 | 0 | 1 | 0 | 0 |
| 0 | 0 | 1 | 1 | 1 |
| 0 | 1 | 0 | 0 | 0 |
| 0 | 1 | 0 | 1 | 0 |
| 0 | 1 | 1 | 0 | 1 |
| 0 | 1 | 1 | 1 | 1 |
| 1 | 0 | 0 | 0 | 0 |
| 1 | 0 | 0 | 1 | 1 |
| 1 | 0 | 1 | 0 | 0 |
| 1 | 0 | 1 | 1 | 1 |
| 1 | 1 | 0 | 0 | 1 |
| 1 | 1 | 0 | 1 | 1 |
| 1 | 1 | 1 | 0 | 1 |
| 1 | 1 | 1 | 1 | 1 |

Q12 Draw a truth table to prove each of the following laws of Boolean Algebra.

- (a) $A + \bar{A} = 1$
 (b) $A + \bar{A} \cdot B = A + B$
 (c) $A \cdot (B + C) = A \cdot B + A \cdot C$

A12 (a)

| A | \bar{A} | $A + \bar{A}$ |
|---|-----------|---------------|
| 0 | 1 | 1 |
| 1 | 0 | 1 |

Thus the result is always 1.
 (b)

| A | B | \bar{A} | $\bar{A} \cdot B$ | $A + \bar{A} \cdot B$ | $A + B$ |
|---|---|-----------|-------------------|-----------------------|---------|
| 0 | 0 | 1 | 0 | 0 | 0 |
| 0 | 1 | 1 | 1 | 1 | 1 |
| 1 | 0 | 0 | 0 | 1 | 1 |
| 1 | 1 | 0 | 0 | 1 | 1 |

The last two columns are identical; thus the equality is proved.

(c)

| A | B | C | $B + C$ | $A \cdot (B + C)$ | $A \cdot B$ | $A \cdot C$ | $A \cdot B + A \cdot C$ |
|---|---|---|---------|-------------------|-------------|-------------|-------------------------|
| 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 |
| 0 | 0 | 1 | 1 | 0 | 0 | 0 | 0 |
| 0 | 1 | 0 | 1 | 0 | 0 | 0 | 0 |
| 0 | 1 | 1 | 1 | 0 | 0 | 0 | 0 |
| 1 | 0 | 0 | 0 | 0 | 0 | 0 | 0 |
| 1 | 0 | 1 | 1 | 1 | 0 | 1 | 1 |
| 1 | 1 | 0 | 1 | 1 | 1 | 0 | 1 |
| 1 | 1 | 1 | 1 | 1 | 1 | 1 | 1 |

Column 5 and column 8 are identical; thus, the equality is proved.

Q13 Construct the logic circuit which will satisfy the following truth table:

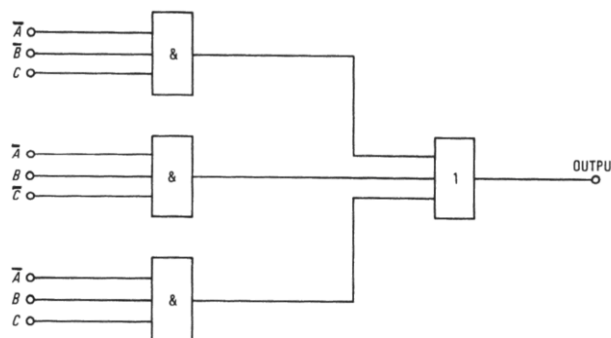
| A | B | C | Output |
|---|---|---|--------|
| 0 | 0 | 0 | 0 |
| 0 | 0 | 1 | 1 |
| 0 | 1 | 0 | 1 |
| 0 | 1 | 1 | 1 |
| 1 | 0 | 0 | 0 |
| 1 | 0 | 1 | 0 |
| 1 | 1 | 0 | 0 |
| 1 | 1 | 1 | 0 |

(6 min)

A13 The basic logic circuit has three elements—one for each of the lines of the truth table which includes logic 1 in the output:

- (a) $\bar{A} \cdot \bar{B} \cdot C$
 (b) $\bar{A} \cdot B \cdot \bar{C}$
 (c) $\bar{A} \cdot B \cdot C$

The logic expression is therefore $\bar{A} \cdot \bar{B} \cdot C + \bar{A} \cdot B \cdot \bar{C} + \bar{A} \cdot B \cdot C$. This can be represented by the logic circuit shown in the sketch.



[Tutorial Note: The logic circuit shown could be simplified, but is left as the basic circuit here for the purpose of illustrating the method of producing a circuit from the truth table.]

Q14 Use Boolean algebra to simplify the following expression:

$$A \cdot B \cdot C + A \cdot \bar{B} \cdot C + \bar{A} \cdot B + \bar{A} \cdot \bar{B} \cdot \bar{C} + \bar{A} \cdot \bar{B} \cdot C \quad (5 \text{ min})$$

A14 The expression can be simplified by first grouping terms.

Thus, $A \cdot B \cdot C + A \cdot \bar{B} \cdot C + \bar{A} \cdot B + \bar{A} \cdot \bar{B} \cdot \bar{C} + \bar{A} \cdot \bar{B} \cdot C$

becomes $A \cdot C \cdot (B + \bar{B}) + \bar{A} \cdot B + \bar{A} \cdot \bar{B} \cdot (\bar{C} + C)$,

which can be reduced to

$$\begin{aligned} & A \cdot C + \bar{A} \cdot B + \bar{A} \cdot \bar{B} \\ &= A \cdot C + \bar{A}, \\ &= \underline{C + \bar{A}}. \end{aligned}$$

Q15 Simplify the logic circuit shown in Fig. 2.

(6 min)

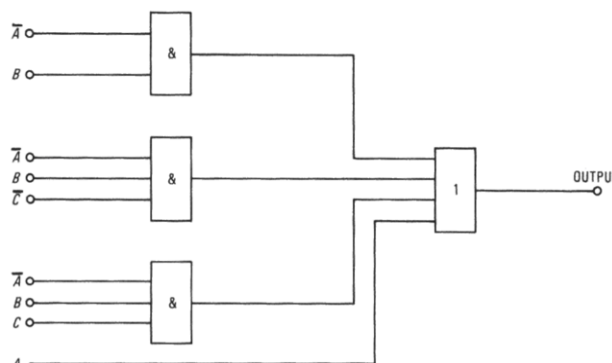


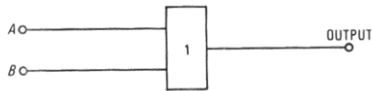
Fig. 2

A15 First, the Boolean expression for the circuit is written down by considering each gate in turn, and then combining the terms for the complete representation.

The expression becomes

$$\begin{aligned} \bar{A} \cdot B + \bar{A} \cdot B \cdot \bar{C} + \bar{A} \cdot B \cdot C + A \\ = \bar{A} \cdot B + \bar{A} \cdot B \cdot (\bar{C} + C) + A, \\ = \bar{A} \cdot B + A, \\ = B + A. \end{aligned}$$

Thus the simplified logic circuit requires only one OR gate, as shown in the sketch.



Q16 Derive the truth table for the logic circuit shown in Fig. 3. (4 min)

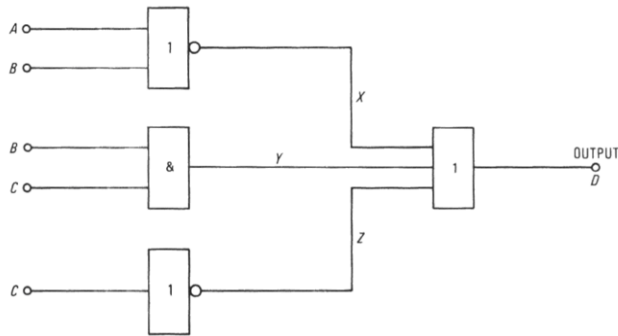


Fig. 3

A16 The truth table can be obtained by considering each circuit element separately.

| A | B | C | X | Y | Z | D = X + Y + Z |
|---|---|---|---|---|---|---------------|
| 0 | 0 | 0 | 1 | 0 | 1 | 1 |
| 0 | 0 | 1 | 1 | 0 | 0 | 1 |
| 0 | 1 | 0 | 0 | 0 | 1 | 1 |
| 0 | 1 | 1 | 0 | 1 | 0 | 1 |
| 1 | 0 | 0 | 0 | 0 | 1 | 1 |
| 1 | 0 | 1 | 0 | 0 | 0 | 0 |
| 1 | 1 | 0 | 0 | 0 | 1 | 1 |
| 1 | 1 | 1 | 0 | 1 | 0 | 1 |

Q17 Name the gate that has the following truth table.

| A | B | C | Output |
|---|---|---|--------|
| 0 | 0 | 0 | 1 |
| 0 | 0 | 1 | 0 |
| 0 | 1 | 0 | 0 |
| 0 | 1 | 1 | 0 |
| 1 | 0 | 0 | 0 |
| 1 | 0 | 1 | 0 |
| 1 | 1 | 0 | 0 |
| 1 | 1 | 1 | 0 |

(1 min)

A17 The gate is a three-input NOR gate.

Q18 A machine tool has three safety interlocks which generate the following logic levels:

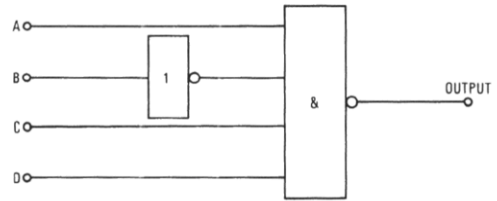
A—1 when locked
B—0 when locked
C—1 when locked

Another input D is the master ON/OFF switch which can operate only when all the interlocks are locked, and it then controls the machine with a logic 0 to turn it on and a logic 1 to turn it off. Derive a suitable logic circuit to control the machine. (5 min)

A18 The logic expression which controls the machine can be written as

$$\bar{A} \cdot \bar{B} \cdot C \cdot D$$

This requires the circuit shown in the sketch.



Q19 Which of the circuits shown in Fig. 4 is equivalent to an OR gate? (6 min)

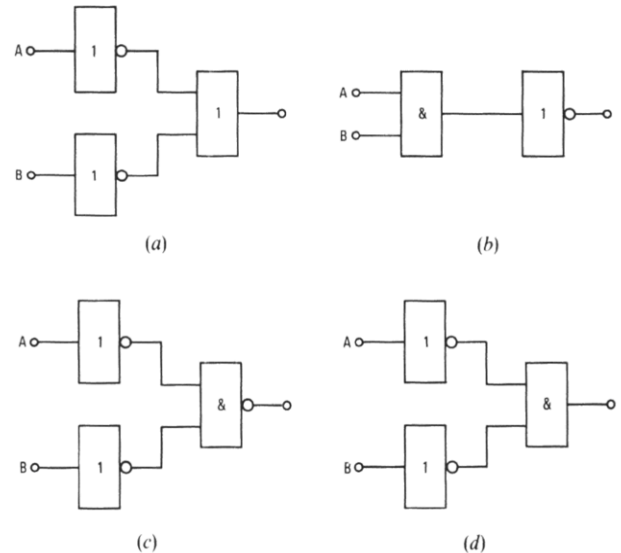


Fig. 4

A19 The equivalent circuit can be obtained by writing down the truth table for each one.

(a)

| A | B | \bar{A} | \bar{B} | Output |
|---|---|-----------|-----------|--------|
| 0 | 0 | 1 | 1 | 1 |
| 0 | 1 | 1 | 0 | 1 |
| 1 | 0 | 0 | 1 | 1 |
| 1 | 1 | 0 | 0 | 0 |

(b)

| A | B | Output |
|---|---|--------|
| 0 | 0 | 1 |
| 0 | 1 | 1 |
| 1 | 0 | 1 |
| 1 | 1 | 0 |

(c)

| A | B | \bar{A} | \bar{B} | Output |
|---|---|-----------|-----------|--------|
| 0 | 0 | 1 | 1 | 0 |
| 0 | 1 | 1 | 0 | 1 |
| 1 | 0 | 0 | 1 | 1 |
| 1 | 1 | 0 | 0 | 1 |

(d)

| A | B | \bar{A} | \bar{B} | Output |
|---|---|-----------|-----------|--------|
| 0 | 0 | 1 | 1 | 1 |
| 0 | 1 | 1 | 0 | 0 |
| 1 | 0 | 0 | 1 | 0 |
| 1 | 1 | 0 | 0 | 0 |

Thus, circuit (c) gives the output equivalent to an OR gate.

Questions and answers contributed by D. Turner

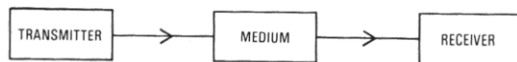
BTEC: TELECOMMUNICATIONS SYSTEMS I

The following questions are based on the BTEC's unit U81/748. Students are advised to read the notes on p. 25

Q1 (a) Draw and label a block diagram (containing three blocks) of a simple communications system.

(b) State three media that can be used for the transmission of information by electrical means. Give an example of a communication system that would use each media stated. (5 min)

A1 (a)



(b)

| Media | System |
|-----------------|-----------|
| Cables or wires | Telephony |
| Free space | Radio |
| Optical fibre | Telephony |

Q2 State the meaning of the following terms; clearly indicate to which type of signal they refer, AC or DC:

- velocity,
- amplitude,
- period, and
- complex.

(5 min)

A2 (a) (i) *Velocity* is the speed at which a signal travels through a medium; for example, along a telephone line or through space. The term applies to both AC and DC signals.

(ii) *Amplitude* is the amount by which the signal deviates from its mean value. The term applies to AC and DC signals.

(iii) *Period* is the length of time a wave takes to complete one cycle. The term applies to AC signals only.

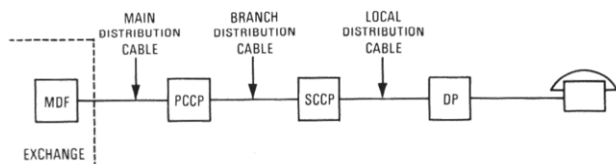
(iv) A *complex* wave is one that is made up of more than one frequency, such as the sound wave produced by the human voice. It contains other frequencies called harmonics. The term applies to AC signals only

Q3 (a) Draw and label a block diagram of a local telephone cable distribution network, including at least one of each of the following:

- secondary cross-connection point (SCCP),
- main distribution frame (MDF),
- distribution point (DP), and
- primary cross-connection point (PCCP)

(b) Briefly explain the function of the items listed. Also explain the main difference between the cables linking them together. (15 min)

A3 (a)



(b) Each of the items provides flexibility between the incoming cables and the outgoing cables. This means that the cables entering any of these items will have a larger capacity and be smaller in number than the cables leaving that item.

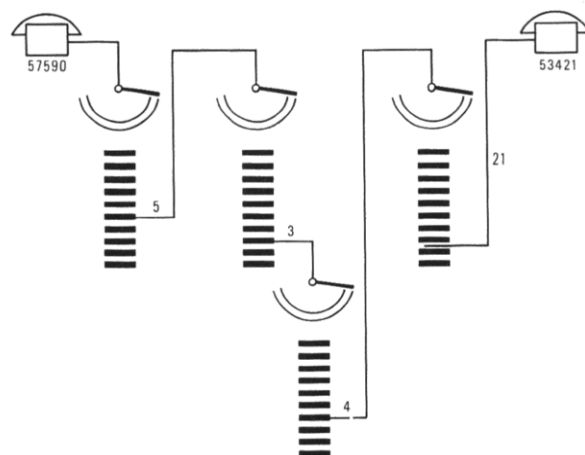
Q4 (a) Draw and label a simple trunking diagram showing a local call in a non-director Strowger telephone exchange linking for following two customers:

Calling customer's number 57590
Called customer's number 53421

(b) State the purpose of each of the supervisory tones that could be used in the local call in part (a).

List the tones that are available at each of the switching stages shown on your diagram. (20 min)

A4 (a)



(b) Dial tone is used to advise the caller that the exchange equipment is ready to accept dial instructions.

Number-unobtainable (NU) tone is used to advise the caller that the number dialled is not available.

Busy tone is used to advise the caller that the number dialled is engaged.

Ring tone is used to advise the caller that the number dialled is being rung.

Equipment-busy tone is used to advise the caller that the equipment at the exchange is busy and to try again later.

Dial tone is available only at the first switching stage. NU, busy and equipment-busy tones are all available at each of the switching stages, including the final selector stage. Ringing tone is available only at the final selector stage.

Q5 Explain what is meant by the following terms when applied to telephone exchanges. Give examples of each type and state the type of switch used in each of them:

- common control,
- step-by-step control.

(10 min)

A5 (a) Common control means that the setting up of the call in the exchange is controlled centrally; therefore, all the dialled digits must be received before the call path can start to be set up. It is used in crossbar and electronic reed-relay exchanges and uses matrix switches.

(b) Step-by-step control means that as each dialled digit is received the next part of the call path can be set up; that is, step-by-step. It is used in Strowger exchanges, both director and non-director, and uses two-motion selector switches.

Q6 A matrix switch has 15 outlets and can carry a maximum of 10 simultaneous calls. Determine

- the total number of inlets on the switch, and
- the total number of crosspoints on the switch.

(5 min)

A6 (a) The total number of simultaneous calls is limited by either the number of inlets or the number of outlets whichever is the lowest; thus, since the maximum number of simultaneous calls is 10, there must be 10 inlets.

(b) Total number of crosspoints

= number of inlets \times number of outlets,

= 10×15 ,

= 150 crosspoints.

Q7 The telephony network is used to connect a central computer with a distant terminal. State:

(a) the type of transmission used, and

(b) the name and function of any special equipment necessary to allow this transmission over the network. (5 min)

A7 (a) Data transmission.

(b) A modem is required at each end of the link. These convert the digital signals from the computer into the analogue signals for transmission over the telephony network and back into digital signals at the distant end.

Q8 Briefly describe each of the following computer peripheral devices; state whether they are input or output devices:

- (a) teletype,
- (b) punch-card reader,
- (c) magnetic-tape reader, and
- (d) visual display unit.

(10 min)

A8 (a) *Teletype* This is like a typewriter, gives a paper copy and is used as an input and output device.

(b) *Punched-Card Reader* Information can be stored as a series of holes on a firm card. The punched-card reader is used to machine read the cards to input the information into a computer. It is thus an input device.

(c) *Magnetic-Tape Reader* This is similar to a punched-card reader in terms of its use, except that the information is stored in magnetic form on reels of tape coated with a magnetic medium.

(d) *Visual Display Unit* This is a television-like device that displays, on a screen, information input to and output from a computer. It is associated with a typewriter-like keyboard for inputting the information.

Q9 List and describe the three main types of radio wave paths and the frequency bands associated with them. (6 min)

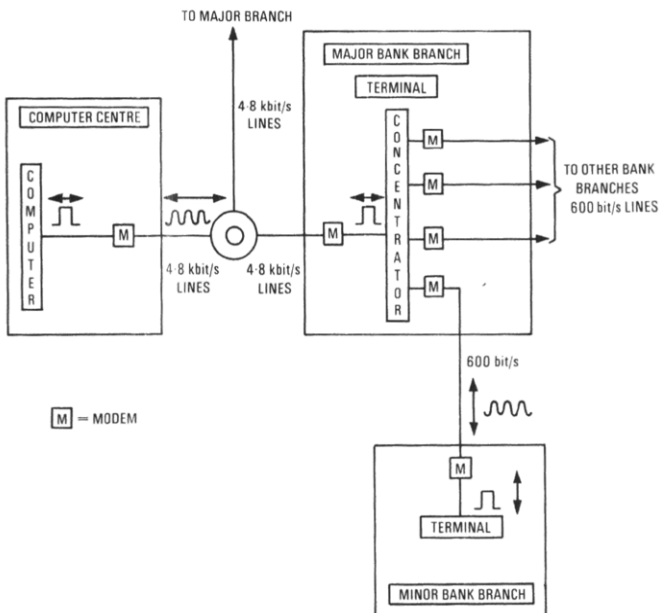
A9 (a) *Ground or Surface Waves* 10 kHz–300 kHz. These waves travel along the surface of the ground and follow the curvature of the earth's surface.

(b) *Sky or Ionospheric Waves* 3 MHz–30 MHz. These waves travel in straight lines into space until they reach the ionosphere, where they are bent until many of them return back to a different part of the earth's surface.

(c) *Space Waves* 30 MHz and above. These also travel in straight lines and either go through the ionosphere or travel over the ground, again in straight lines. Reception is by line of sight and so is limited by the curvature of the earth's surface.

Q10 Draw and label a block diagram of a typical bank's data-communication system and briefly explain its operation. Include in your diagram the central bank computer headquarters, two major bank branches, one of which serves four minor branches. Each branch has on-line facilities and can work in real time. (20 min)

A10



The central computer can deal only with one terminal at a time but, because it works very quickly, it can deal with many branches. It uses a polling technique whereby it addresses each branch individually and in rotation to see if it has any information to exchange. When it completes one exchange, it then moves on to the next branch. It does all this extremely quickly.

Q11 Draw and label a block diagram of a commercial broadcast system. Include in your answer at least one cable link. State one advantage for including this link. (5 min)

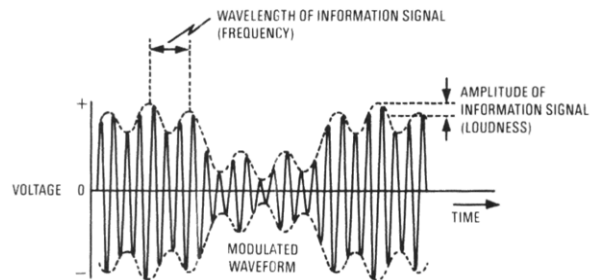
A11



The cable link would be preferred because it is more secure and requires less equipment; it is therefore cheaper than using an equivalent radio link.

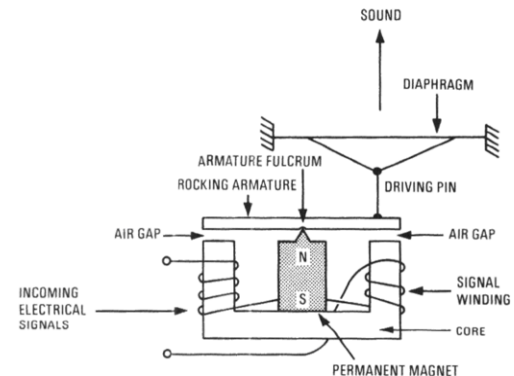
Q12 Draw and label a waveform diagram of an amplitude-modulated wave. Identify on your diagram those components which convey the loudness and the frequency of the information signal. (10 min)

A12



Q13 Explain, with the aid of a labelled sketch, how sound is produced by the rocking-armature receiver. (20 min)

A13

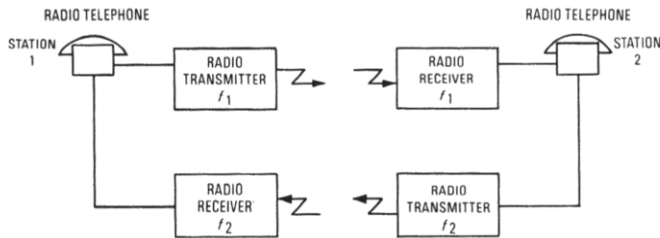


The magnetic field produced by the permanent magnet at the centre spreads through the two outer limbs and back to the opposite pole of the magnet through the air gaps, the armature and the central pivot. The sketch shows that the armature can move about the pivot in a see-saw manner; hence the name *rocking armature*. With no current flowing in the signal coils, the armature takes up a balanced position because the permanent magnet field is of equal strength in each pole gap. The signal windings are each wound on an outer limb, but in opposite directions so that when current passes through them a north or south pole is produced. When the signal current reverses, the polarities of the coils also reverse.

When the signal current is in one direction, the magnetic field produced adds to the permanent field in one gap, and weakens the field in the other. This results in the armature tilting towards the pole with the strongest field at that instant and hence either pulling or pushing the diaphragm from the normal position, depending upon which way the armature has been tilted. When the direction of the signal current is reversed, the opposite is the case. For all signal current values, either in one direction or the other, the very light armature moves in sympathy and in so doing moves the diaphragm to produce sound.

Q14 Draw a block diagram of a radio-telephone system, clearly showing, in general terms, the radio channels used. Explain why these channels are used. (10 min)

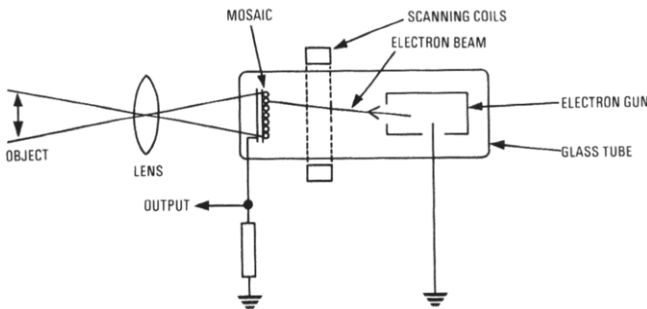
A14



For the link to be two-way, without physical switching, the frequencies f_1 and f_2 must be different. This ensures that station 1's transmission frequency is not received by station 1's receiver; similarly for station 2.

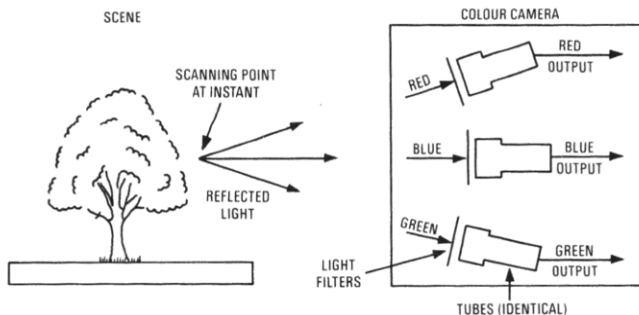
Q15 With the aid of a simple labelled sketch, explain the principle of operation of a black-and-white television camera. Explain how this principle is used for a colour camera. (20 min)

A15 Sketch (a) shows the principle of operation of a black-and-white television camera. The evacuated glass tube contains a photo-emissive mosaic and an electron gun. The electron beam produced by the gun is directed onto the mosaic, which it scans. Movement of the beam is given by the scanning coils which, although located on the outside of the tube, influence the electrons by their magnetic effect through the tube glass. Although this type of camera is no longer used, the principle of operation is still the basis for present-day camera tubes.



(a)

See sketch (b). In a colour television camera, there are three tubes, each of which responds to the brightness of one of the primary colours in the scene. This is done with the aid of filters. Thus, three outputs, one for each colour, are produced and electronically combined to produce a colour picture and the receiver.



(b)

Q16 The number of picture lines used for ultra-high-frequency (UHF) television in the UK is

- (a) 405,
- (b) 525,
- (c) 625, or
- (d) 819.

(2 min)

A16 (c) 625

Q17 The number of interlaced pictures that make up the completed television picture is

- (a) 2,
- (b) 4,
- (c) 25, or
- (d) 50.

(2 min)

A17 (a) 2

Q18 State the two methods of accessing information from the Prestel computer database, and briefly describe each method. (10 min)

A18 The two methods are:

- (a) the index search method, and
- (b) the directly-keyed method.

In the index search method, the user is first presented with a general index, usually 1–10. From this, the number of the general heading required is keyed. The user is then presented with another index relating only to the general heading just nominated. The user must then key 1–10 for the general section of the main heading and so on. This process is repeated until the required page is reached.

In the directly-keyed method, as the name suggests, the user must key in the code for the particular page required. The user can look up this code in a directory before calling the Prestel computer, thus saving a great deal of user and computer time.

Q19 A typical ultra-high-frequency broadcast frequency is

- (a) 615 kHz,
- (b) 45 MHz,
- (c) 615 MHz, and
- (d) 45 GHz.

(2 min)

A19 (c) 615 MHz.

Q20 Primary and secondary radar are two forms of navigation systems.

- (a) Give a simple explanation of each of these systems, and
- (b) State the main difference between these two systems in terms of frequency of operation, system power and data capability.

A20 (a) Primary radar is normally carried by both ships and aircraft and makes use of the fact that radio waves are reflected by objects such as a land mass, buildings, buoys, other ships etc. The reflected wave is used to fix the position of such objects relative to the source of the wave itself.

Secondary radar is a very strong signal source permanently positioned and giving its identification to any radar receiver coming into its area. These stations are commonly referred to as *beacons*.

(b) (i) Frequency

Primary radar installations normally operate at fixed frequencies in a band ranging approximately from 3–10 GHz.

Secondary radar, therefore, must be able to respond to an input signal having any frequency in the above band. It must also return a signal at the same frequency as the interrogating radar and, in order to do this, its transponder, when activated by an incoming signal, transmits sequentially, or sweeps, through the whole frequency band at intervals of approximately two minutes.

(ii) System Power

Much greater power is needed in a primary radar system than in a secondary system because, in the former, the signal has to travel out to the target and return as an echo to the receiver. In a secondary system, only half the distance is involved, which is that between secondary responder and primary receiver.

Typical values are:

Primary radar—0.25–1.25 kW input power, depending upon size, up to 30 kW output power in concentrated bursts.

Secondary radar—1–2 W input power, but can be as low as one eighth of a watt when not transmitting. Up to 10 kW output power.

(iii) Data Capability

Secondary radar systems can identify themselves by displaying a code as well as marking their positions on a plan position indicator.

Q21 State the type of signal produced by a teleprinter and whether the signal is digital or analogue. (4 min)

A21 A teleprinter produces a double-current signal, which is a digital signal.

Questions and answers contributed by R. L. Odell

BTEC: RADIO III

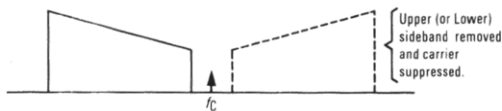
The questions in this paper are based on the BTEC's standard unit U81/744. Students are advised to read the notes on p. 25

Q1 (a) With reference to radio communication systems using amplitude modulation, explain the terms

- (i) SSB, and
- (ii) ISB systems.
- (b) State ONE use of SSB systems and ONE use of ISB systems.
- (c) Briefly explain the disadvantages of suppressed carrier transmission. (20 min)

A1 (a) (i) SSB

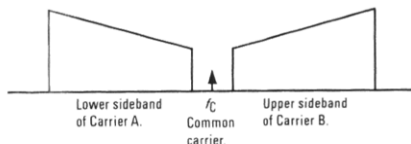
Single-sideband suppressed-carrier amplitude modulation is frequently known as SSB. Since both sidebands carry the information, it is not necessary to transmit them both; thus if only one sideband is transmitted, the bandwidth of the system is halved. In such a system, it is also usual to suppress the carrier, and the system is then said to be operating in *single-sideband suppressed-carrier mode* (see sketch (a)).



(a)

(ii) ISB

An alternative system to the SSB system is the independent sideband (ISB) system. In such a system, two signals are used to modulate two carriers derived from the same oscillator. The signal that is transmitted consists of the upper sideband of one carrier and the lower sideband of the other. Thus, two sets of information can be made to occupy the same bandwidth as a normal double-sideband (DSB) transmission (see sketch (b)).

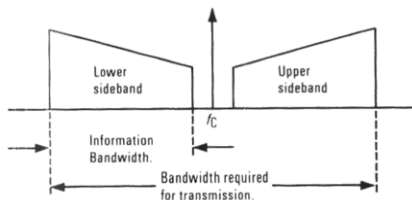


(b)

(b) SSB transmission is usually used in short-wave very-high-frequency (VHF) communication systems.

ISB is used for telephone communications systems.

(c) The main disadvantage of suppressed-carrier transmissions is that, because the carrier is transmitted at a much lower level than the sidebands (for example, 27 dB down), the receiver must contain an oscillator working at the carrier frequency so that the carrier component can be re-inserted into the received signal and the received signal demodulated to produce the correct information. This oscillator must operate at the same frequency as the oscillator in the transmitter (and in some cases must also be in phase with the transmitter to avoid distortion). This makes the receiver very costly and not an economical proposition for normal entertainment broadcasts. Thus, radio transmitters used for broadcasting sound in long- and medium-wave bands radiate both sidebands and carrier (see sketch (c)).



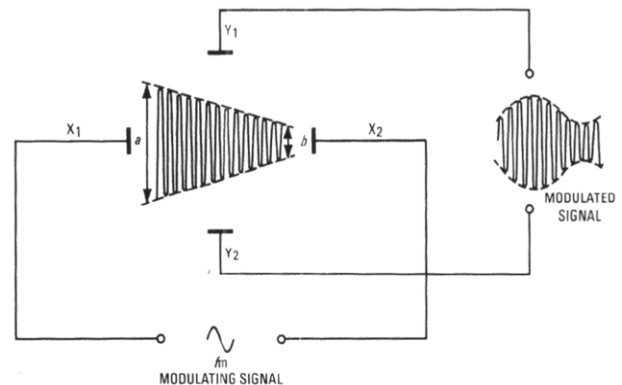
(c)

Q2 Describe a suitable method of measuring the depth of modulation of an amplitude-modulated (AM) signal. (15 min)

A2 In the trapezoidal method, illustrated in sketch (a), the modulated carrier signal is applied to the Y plates of a cathode-ray oscilloscope (CRO), and the modulating signal to the X plates, with the CRO time-base switched off.

When the modulating signal is at a maximum, the spot is deflected over to the left-hand side of the screen; this coincides with the amplitude-modulated signal being at a maximum, and gives the displacement a .

When the modulating signal is at a minimum, the spot is over towards the right-hand side of the screen; this coincides with the troughs of the amplitude-modulated signal and gives the displacement b .

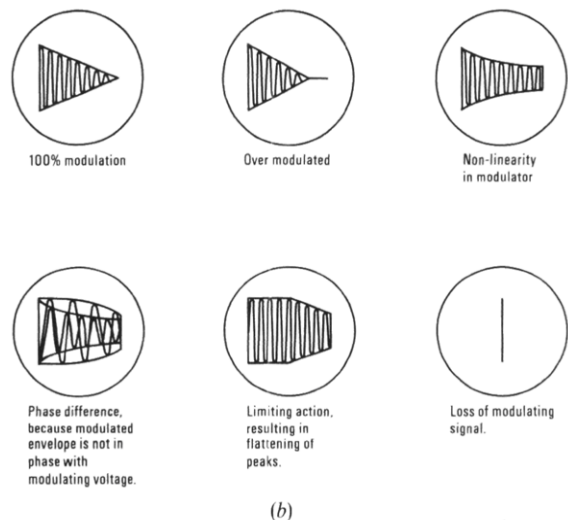


(a)

The depth of modulation, m , is given by

$$m = \frac{a - b}{a + b}$$

[Tutorial note: In this method, the measurement of a and b is simple, as shown in sketch (a), and several types of carrier distortion can easily be recognised, as shown in sketch (b).]



(b)

Q3 (a) Briefly explain the following terms in connection with frequency-modulation (FM) systems:

- (i) frequency deviation,
- (ii) modulation index, and
- (iii) practical bandwidth.

(b) State the rated system deviation used in FM radio broadcasts in the UK.

(c) When the modulation index of a certain FM transmitter is 5 in a practical bandwidth of 180 kHz, what is its frequency deviation? (20 min)

A3 (a) (i) The difference between the instantaneous modulated frequency and the unmodulated frequency is called the *frequency deviation*, δ_f , of the modulated signal. It is proportional to the amplitude of the modulating signal.

(ii) The modulation index, m , can be defined as the ratio of the frequency deviation, δ_f , to the frequency of the modulating signal, f_m ; that is, $m = \delta_f / f_m$.

(iii) The practical bandwidth, $B = 2(\delta_f + f_m)$.

(b) ± 75 kHz.

[Tutorial note: Television sound: ± 50 kHz.]

- (b) (i) Let f_{sm} = maximum signal frequency,
 f_{sr} = required signal frequency,
 C_{sm} = capacitance at maximum signal frequency, and
 C_{sr} = capacitance at required signal frequency.

Now,

$$f_{sm} = \frac{1}{2\pi\sqrt{LC_{sm}}}, \text{ and} \quad \dots\dots(1)$$

$$f_{sr} = \frac{1}{2\pi\sqrt{LC_{sr}}}. \quad \dots\dots(2)$$

[Tutorial note: In the aerial tuned circuit, the resonant frequency, f , in hertz is given by:

$$f = \frac{1}{2\pi\sqrt{LC}},$$

where L is the inductance in henrys and C is the capacitance in farads. The maximum frequency corresponds to minimum capacitance.]

Dividing equation (1) by equation (2) gives

$$\frac{f_{sm}}{f_{sr}} = \frac{2\pi\sqrt{LC_{sr}}}{2\pi\sqrt{LC_{sm}}} = \sqrt{\left(\frac{C_{sr}}{C_{sm}}\right)}.$$

$$\therefore f_{sr} = \frac{f_{sm}}{\sqrt{\left(\frac{C_{sr}}{C_{sm}}\right)}}.$$

Substituting the given values gives

$$f_{sr} = \frac{1500 \times 10^3}{\sqrt{\left(\frac{220 \times 10^{-12}}{40 \times 10^{-12}}\right)}} = \frac{1500 \times 10^3}{2.345} \text{ Hz},$$

$$= 639.66 \text{ kHz}.$$

- (ii) Let f_{lom} = maximum local-oscillator frequency,
 f_{lor} = required local-oscillator frequency,
 C_{lom} = capacitance at maximum local-oscillator frequency,
and
 C_{lor} = capacitance at required local-oscillator frequency.

It is assumed that the local-oscillator frequency is the sum of the signal and intermediate frequencies.

Now,

$$f_{lor} = \frac{f_{lom}}{\sqrt{\left(\frac{C_{lor}}{C_{lom}}\right)}}.$$

Substituting the given values gives

$$f_{lor} = \frac{1965 \times 10^3}{\sqrt{\left(\frac{290 \times 10^{-12}}{110 \times 10^{-12}}\right)}} = \frac{1965 \times 10^3}{1.624},$$

$$= 1209.98 \text{ kHz}.$$

[Tutorial note: f_{sr} and f_{lor} have been derived in a similar manner, see equation (1).]

- (iii) The mid-point of the capacitance range corresponds to the values calculated in (i) and (ii).

Therefore, the tracking error

$$= 1209.98 - (639.66 + 465) = 1209.98 - 1104.66,$$

$$= 105.32 \text{ kHz}.$$

Q8 (a) Briefly explain the term 'noise factor' as applied to radio receiver.

(b) The input to an amplifier consists of 10 mW of noise-free signal together with 1 μW of noise. The power gain of the amplifier is 20 dB. The noise power generated within the amplifier measured at the output is 20 μW.

Calculate

- (i) the signal-to-noise ratio at the input in decibels,
(ii) the signal-to-noise ratio at the output in decibels, and
(iii) the noise factor. (15 min)

A8 (a) Noise factor is defined as the ratio of the signal-to-noise ratio of the input to the signal-to-noise ratio of the output. It is a measure of the

degree to which the receiver degrades the input signal-to-noise ratio.

An ideal receiver, contributing no noise, would have a noise factor of unity.

- (b) The signal-to-noise ratio at the input

$$= 10 \log_{10} \frac{10 \times 10^{-3}}{1 \times 10^{-6}},$$

$$= 10 \log_{10} 10000,$$

$$= 40 \text{ dB}.$$

- (ii) The signal power at the output

$$= \text{gain of the amplifier} \times \text{signal power in},$$

$$= 100 \times 10 \times 10^{-3},$$

$$= 1 \text{ W}.$$

The noise power at the output

$$= 100 \times \text{input noise power}$$

$$+ \text{noise generated by the amplifier},$$

$$= 100 \times 1 \times 10^{-6} + 20 \times 10^{-6},$$

$$= 120 \mu\text{W}.$$

The signal-to-noise ratio at the output

$$= 10 \log_{10} \frac{1}{120 \times 10^{-6}},$$

$$= 10 \log_{10} 8333.3,$$

$$= 39.2 \text{ dB}.$$

- (iii) Noise factor = $\frac{\text{signal-to-noise ratio at input}}{\text{signal-to-noise ratio at output}},$

$$= \frac{10^4}{8.33 \times 10^3} = 1.2.$$

[Tutorial Note: Note that, in the calculation of noise factor, arithmetic ratios are used, not ratios expressed in decibels.]

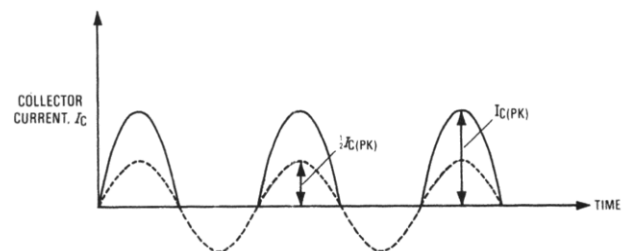
Q9 (a) Why are class-C tuned power amplifiers not normally used for amplifying amplitude-modulated signals?

- (b) Calculate the maximum possible efficiency for class-B amplifiers.

A9 (a) Class-C tuned power amplifiers introduce distortion because the output is not proportional to the input.

[Tutorial note: Class-C tuned power amplifiers are very efficient and are commonly used when the input drive is of constant amplitude.]

(b) The collector current of a class-B tuned amplifier appears as a series of half-sinewave pulses, as shown in the sketch, when a sinusoidal voltage is applied to it.



The fundamental frequency component of this current develops a voltage, V_L , across the collector tuned circuit having a peak value of $\frac{1}{2}I_{c(pk)}$. The mean value of this current is $(1/\pi)I_{c(pk)}$. Harmonics are ignored.

The output power,

$$P_{AC} = V_{RMS} I_{RMS} = \frac{V_L}{\sqrt{2}} \times \frac{I_{c(pk)}}{2\sqrt{2}} = \frac{I_{c(pk)} \times V_L}{4}.$$

But,

$$V_L = V_{cc} - V_c,$$

where V_{cc} = power supply voltage, and
 V_c = collector voltage.

$$\therefore P_{AC} = \frac{I_{c(pk)}(V_{cc} - V_c)}{4}$$

The input power,

$$P_{DC} = \frac{I_{c(pk)}}{\pi} \times V_{cc}$$

Therefore, the collector efficiency

$$\begin{aligned} &= \frac{P_{AC}}{P_{DC}}, \\ &= \frac{I_{c(pk)}(V_{cc} - V_c)}{4} \times \frac{\pi}{I_{c(pk)} \times V_{cc}} \times 100\%, \\ &= \frac{(V_{cc} - V_c)}{4} \times \frac{\pi}{V_{cc}} \times 100\%. \end{aligned}$$

The efficiency depends on how closely the radio-frequency output voltage, V_L , approaches the supply voltage, V_{cc} .

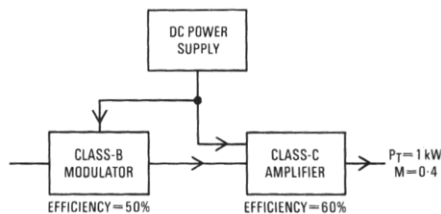
If $V_L = V_{cc}$, then $V_c = 0$ as $V_L = V_{ce} - V_c$.
 Therefore, the maximum collector efficiency

$$= \left(\frac{V_{cc} - 0}{4} \right) \times \frac{\pi}{V_{cc}} \times 100\% = \frac{\pi}{4} \times 100\% = 78.5\%.$$

Q10 A sinusoidally modulated output waveform, from a collector-modulated class-C amplifier, has a depth of modulation of 40% and a total power of 1 kW. The efficiencies of the class-B modulator and class-C amplifier are 50% and 60% respectively. Calculate

- the modulation power supplied to the collector of the final stage,
- the total input power to the final class-C amplifier stage, and
- the DC power supplied to the class-B modulator stage. (15 min)

A10 See sketch.



- It can be shown that the total modulated power (P_T) in watts is given by

$$P_T = P_c \left(1 + \frac{m^2}{2} \right),$$

where P_c is the carrier power in watts, and m is the modulation depth.
 Since $m = 0.4$ and $P_T = 1000$ W,

$$1000 = P_c \left(1 + \frac{0.4^2}{2} \right) = P_c \left(1 + \frac{0.16}{2} \right) = 1.08 P_c.$$

$$\therefore P_c = \frac{1000}{1.08} = 925.9 \text{ W}.$$

The total modulated power (P_T) is the sum of the sideband power output, P_s , and the carrier power output, P_c .

$$\therefore P_s = 1000 - 925.9 = 74.1 \text{ W}.$$

Thus the modulating power input to the final stage

$$= \frac{74.1}{0.6} = 123.5 \text{ W},$$

(as the efficiency of the final stage is 60%).

- DC power input to the final stage

$$= \frac{\text{carrier power output}}{\text{amplifier efficiency}},$$

$$\begin{aligned} &= \frac{925.9}{0.6}, \\ &= 1543.2 \text{ W}. \end{aligned}$$

Total input power to the final-stage collector

$$\begin{aligned} &= 1543.2 + 123.5, \\ &= 1666.7 \text{ W}. \end{aligned}$$

- As shown in (a), the sideband power supplied to the class-C stage is 123.5 W.

Therefore, the DC power supplied to the class-B modulator stage

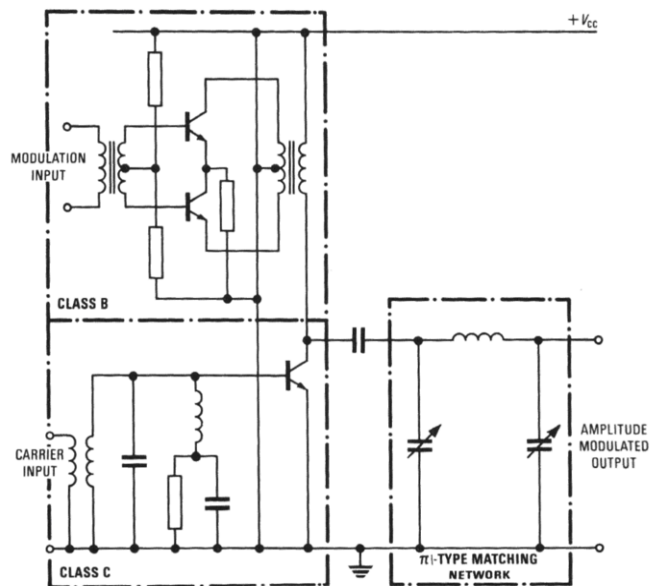
$$= \frac{123.5}{0.5} = 247 \text{ W},$$

(since the class-B modulator stage has an efficiency of 50%).

[Tutorial note: The carrier power output is equal to the DC power input to the final stage multiplied by the efficiency of the final stage; the sideband power is equal to the DC power input to the modulating stage multiplied by the product of the efficiencies of the modulator and final stages.]

- Sketch the circuit of a collector-modulated class-C power amplifier with a π -type network to couple the amplitude-modulated output A11 to the aerial. (10 min)

A12



- What is meant by the terms

- reflection coefficient, and

- voltage standing wave ratio,

as used with reference to an aerial and feeder system?

- Briefly explain the important feature of the aerial-coupling system.

- Give an example of a practical application of the use of voltage-standing-wave-ratio metering. (12 min)

A12 (a) (i) A generally convenient method of expressing the state of mismatch on the transmission line is to take the ratio of forward and reflected voltages or currents. The reflection coefficient, ρ , is thus

$$\rho = \frac{\text{magnitude of the reflected wave}}{\text{magnitude of the forward wave}}.$$

[Tutorial note: It can be shown that ρ is determined by the values of the characteristic impedance, Z_0 , and the load impedance, Z_L , of the line, as follows:

$$\rho = \frac{Z_L - Z_0}{Z_L + Z_0} = \frac{Z_L/Z_0 - 1}{Z_L/Z_0 + 1}.$$

(ii) The degree of mismatch on a line can be determined by the ratio of voltage (or current) maximum to minimum values.

The maximum and minimum values can be found by considering the voltage (or current) at any point as the resultant of forward and reflected waves. The voltage standing-wave ratio (VSWR) on the line = V_{\max}/V_{\min} .

[Tutorial note: It can be shown that the

$$\text{VSWR} = \frac{1 + \rho}{1 - \rho}$$

and, by substituting the value of ρ ,

$$\text{VSWR} = \frac{Z_L}{Z_0}$$

(b) The important feature of an aerial-coupling system is that there should be a good impedance match throughout. In other words, the output impedance of the transmitter, the characteristic impedance of the feeder and the input impedance of the aerial should ideally be equal.

The impedance mismatch between the transmitter and the aerial sets up a standing wave on the line; this is, the voltage and current along the feeder are not constant but vary along the length of the line. The presence of standing waves on a feeder indicates loss of power. Excessive

VSWR is not permissible as this can cause breakdown of coaxial cable due to high voltage and can damage the transmitter output stage.

(c) The arrangement shown in sketch (a) can be used to provide best power matching. The matching unit is adjusted for best results.

The arrangement shown in sketch (b) can be used for protection. The VSWR trip circuit can be designed to operate on the transmitter when the VSWR reaches the value of, say, 1.1.

Q13 Complete the following table for Yagi, log-periodic and rhombic aerials as applied to a typical aerial of each type. (10 min)

| | Yagi | Log-Periodic | Rhombic |
|---------------------------|------|--------------|---------|
| Operating frequency range | | | |
| Gain | | | |
| Input impedance | | | |
| Common use | | | |

A13

| | Yagi | Log-Periodic | Rhombic |
|---------------------------|--|---------------------------------|-----------------------------|
| Operating frequency range | 88 MHz–850 MHz | 450 MHz–850 MHz | 3 MHz–200 MHz |
| Gain | 5–30 dB | 9 dB | 7–15 dB |
| Input impedance | 73 Ω | 300 Ω | 600 Ω |
| Common use | Reception of radio and television in VHF and UHF bands | Reception of television signals | High-frequency transmission |

Questions and answers contributed by P. R. Das

CITY AND GUILDS OF LONDON INSTITUTE Telecommunications Technicians (New) Scheme

The following questions are from examination papers set for the City and Guilds of London Institute's (CGLI's) new 271 Telecommunications Technicians Scheme, and are reproduced with the permission of the CGLI. The answers given have been prepared by independent authors. Answers to some questions are occasionally omitted because of insufficient space. Students studying BTEC courses at the higher level may find that these questions are useful for revision.

CGLI: TRANSMISSION T4 OPTION (1985)

Students are advised to read the notes above. The time for the paper was three hours. Students had to answer six questions

Q1 (a) With reference to a speech signal explain how an audio cable pair introduces

- amplitude/frequency distortion
- phase/frequency distortion
- delay/frequency distortion.

(b) Explain with the aid of simple sketches what is meant by

- near-end crosstalk
- far-end crosstalk.

(c) State **FOUR** sources of crosstalk in a multipair cable.

(d) Discuss **TWO** techniques available for the elimination of crosstalk.

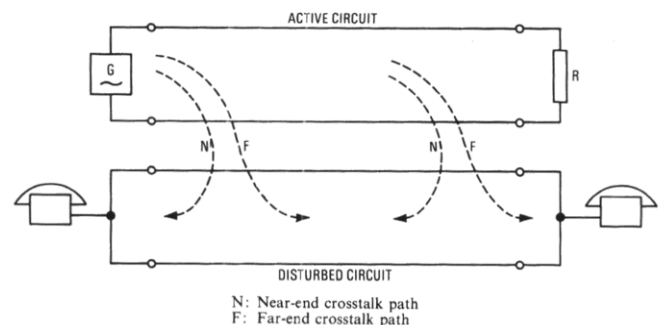
A1 (a) (i) An audio cable attenuates different frequencies by different amounts. A speech signal can consist of many different frequencies, and so these have different amplitudes relative to each other when they have passed along the cable.

(ii) Different frequencies travel at different speeds along the cable, and so, at any instant at the distant end, the different frequencies in the speech signal have different relative phases.

(iii) The rate of change of phase velocity in (ii) is not linear with frequency. Although the human ear is tolerant to this effect, signals are spread out in time at the distant end because of the individual fre-

quencies not all arriving at the same time. This is of more importance in data transmission and can result in adjacent bits overlapping each other.

(b) (i) and (ii) See sketch. The induced signal which appears in the disturbed circuit in a direction opposite to the direction of propagation in the disturbing circuit is known as *near-end crosstalk*. When crosstalk is



propagated in a disturbed channel in the same direction as the direction of propagation in the disturbing channel, it is called *far-end crosstalk*.

(c) Any four of the following:

capacitance unbalance,
resistance unbalance,
inductive coupling,
low insulation resistance, or
wire-to-wire contacts.

(d) Any two of the following:

(i) Sound design and the use of good manufacturing techniques to ensure uniformity will minimise some of the factors listed in (c). For example, the main source of crosstalk on carrier systems working over pair-type cable is due to capacitance unbalance. Ideally, all inter-wire capacitances should be the same, as should the capacitances of each wire to earth, but this is rarely so. During manufacture great care is taken to ensure uniformity. The four wires of each quad are taken from the same reel, thus eliminating any difference in diameter due to a drawing die, all are insulated with paper ribbons cut from the same roll and the same amount of ink is used in the coding bars.

(ii) Careful installation. Capacitance unbalance in every cable length can be measured directly and the wires to be jointed can be selected in such a way that the resultant unbalance in the whole cable is kept to a minimum.

(iii) Trimming capacitors can be connected to the cable at the receive terminal to neutralise any remaining unbalance. This is effective in minimising far-end crosstalk only.

To reduce near-end crosstalk, the transmit and receive circuits can be put in separate cables. Screened cable and/or the distance separating the circuits significantly reduce near-end crosstalk.

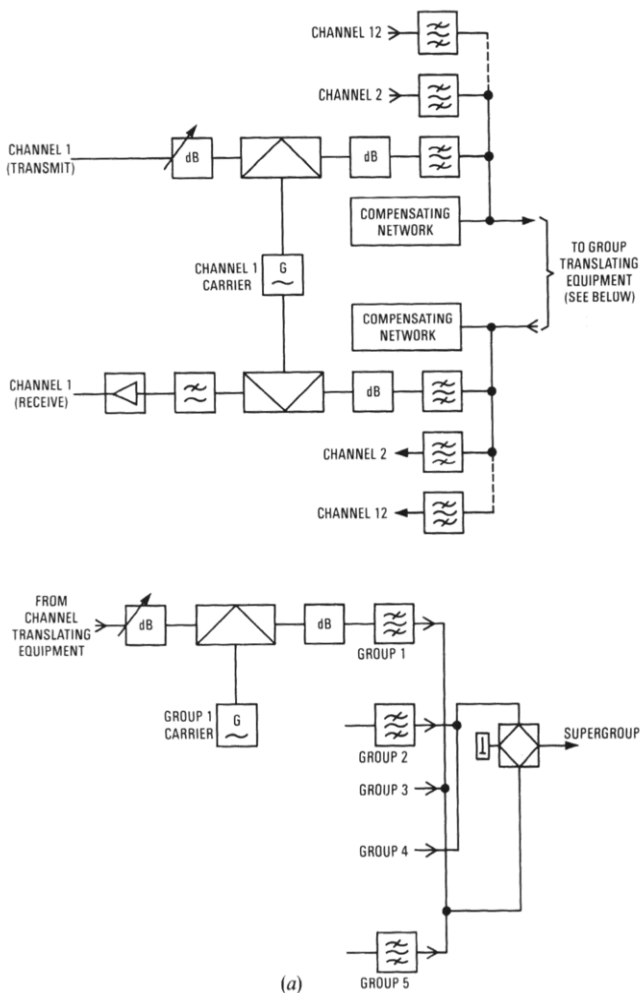
Q2 (a) Draw and fully label a system block diagram to show the formation of a frequency-division multiplexed (FDM) supergroup.

(b) State the carrier frequencies used to form a supergroup.

(c) Draw the assembly block diagram of a hypergroup, stating the number of speech channels available.

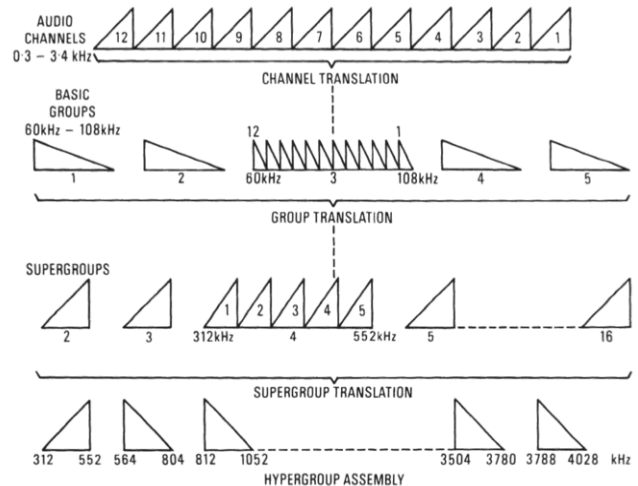
(d) Explain the need for guard bands in the assembly of a hypergroup.

A2 (a) See sketch (a).



(b) 420 kHz, 468 kHz, 516 kHz, 564 kHz, 612 kHz.

(c) See sketch (b). A 15-supergroup hypergroup is assumed.



(b)

The number of speech channels = 900.

(d) Guard bands prevent electrical signals from overlapping which would cause interference on the demodulated channels.

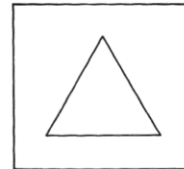
Q3 (a) State the THREE factors necessary to ensure the synchronisation of a transmitter and receiver in a facsimile system.

(b) Explain with the aid of sketches how variations in each characteristic will affect the received print.

(c) Derive an expression to show the relationship between the system bandwidth and the time required to transmit a picture.

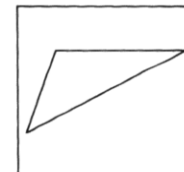
A3 (a) (i) Speed,
(ii) phase, and
(iii) aspect ratio (index of co-operation).

(b) If the transmitted pattern is as follows,



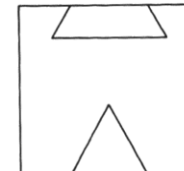
then the variations are as follows:

(i)



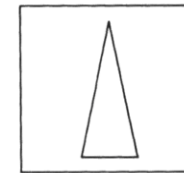
(that is, skewed)

(ii)



(that is, the beginning and end are misaligned)

(iii)



(that is 'stretched' or 'squashed')

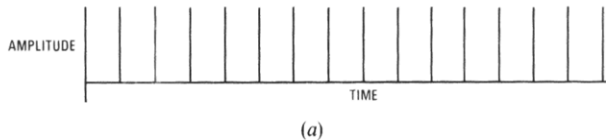
(c) Assuming uncoded transmission, the system bandwidth determines the maximum number of pixels sent per second. If the number of pixels per line is n and the aspect ratio is m , then the number of pixels to be sent is $n \times m$. The maximum information to be transmitted is therefore $nm/2$ (equivalent to alternating black and white pixels). Thus the time taken to transmit is $nm/2B$, where B is the bandwidth in hertz.

Q4 (a) Explain with the aid of simple time-domain sketches, the principle of pulse-amplitude modulation (PAM).

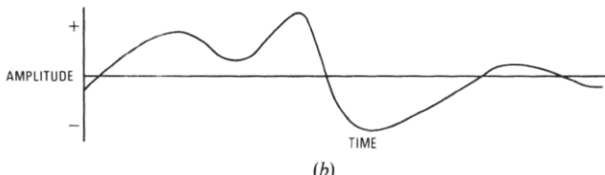
(b) Explain with the aid of PAM frequency spectrum diagrams why the PAM sampling rate must be at least TWICE the highest signal frequency.

(c) Explain why direct PAM transmission is NOT generally used.

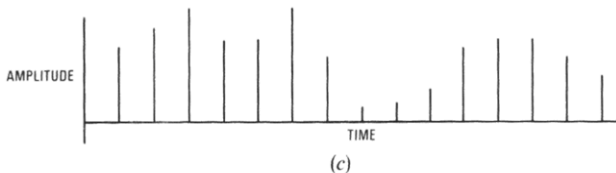
A4 (a) When a sampling-pulse train as shown in sketch (a) is used to sample the complex waveform in sketch (b), a pulse-amplitude-modulation signal (PAM), as shown in sketch (c), results. As can be seen, the pulse heights in sketch (c) are proportional to the amplitude of the waveform shown in sketch (b).



(a)

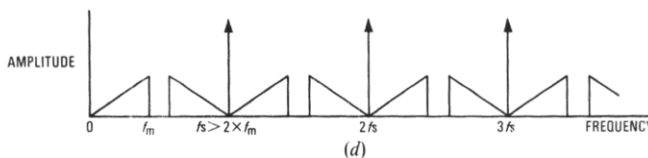


(b)

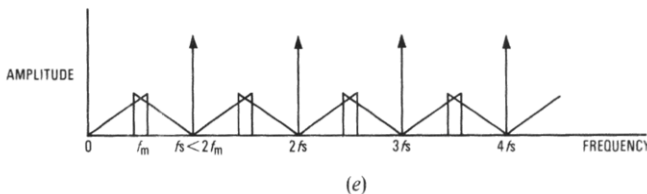


(c)

(b) The frequency spectrum of a PAM signal consists of a series of spectral lines spaced at intervals of f_s , where f_s is the sampling frequency. Each has sidebands, as shown in sketch (d), including an erect sideband on the upper side of zero frequency. In demodulation, this can be extracted by passing the signal through a low-pass filter which cuts off frequencies above f_m , the maximum modulation frequency. Sketch (d) shows a PAM spectrum for a modulating signal with bandwidth equal to f_m and a sampling frequency more than $2 \times f_s$. Sketch (e) shows the effect of using a sampling frequency of less than $2 \times f_m$. As can be seen from sketch (e), the sidebands overlap and distortion occurs in the demodulated signal. Thus, the sampling rate must be at least twice the highest signal frequency.



(d)



(e)

(c) When transmitted to line, the PAM pulses would be subjected to too much distortion for the recovered signal to be useful. Speech has a considerable range of amplitudes and PAM would result in high-amplitude pulses which could introduce interference into other systems.

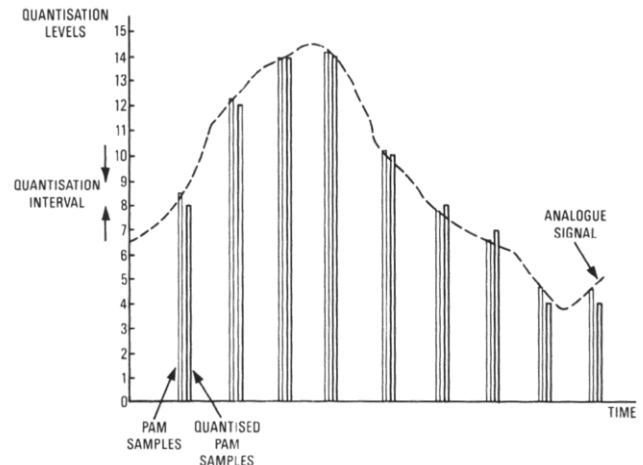
Q5 (a) Explain, with the aid of a simple sketch, the principle of quantisation in a pulse-code modulation (PCM) system.

(b) Distinguish between linear and non-linear quantisation.

(c) Explain the term 'quantisation distortion' and hence explain the need for non-linear quantisation in PCM systems.

A5 (a) The basis of pulse-code modulation (PCM) is to sample a message signal at regular intervals, convert this sample to a series of pulses or code and then reverse the process to recover the original signal at the receiving end.

Consider the sampled, that is, PAM, version of an analogue signal shown in the sketch. Ideally, each PAM pulse should be given a code representing its exact amplitude. However, this is not practicable, as it would be necessary to have a large code to represent each PAM value and a corresponding increase in information transmitted to line. Instead, each pulse is represented by a discrete value or quantum by a process termed *quantisation*. The sketch also shows the quantised version of the PAM signal in the sketch.



(b) The sketch in part (a) shows an example of linear quantisation; that is, the quantum levels are equally spaced. In non-linear quantisation, the quantum intervals are unequal over the input-signal amplitude range.

(c) A result of using a finite number of quantisation levels is that a PAM signal may be allocated a quantising level that is slightly higher or lower than the actual signal level and, hence, some error is introduced to the signal. This error is called *quantisation error* or *distortion* and the effect of a succession of quantisation errors is referred to as *quantisation noise*; the use of more quantisation levels reduces the noise but increases the information to be sent to line.

The use of linear quantisation levels means that the signal-to-quantisation-noise ratio is the same irrespective of the amplitude of the signal. Quieter speech is encoded by fewer quantisation levels than louder speech; this results in greater quantisation distortion on quiet speech and, hence, a worse signal-to-noise ratio. To obtain a signal-to-noise ratio that is almost constant over the whole input-signal range, the intervals between quantisation levels at the lower levels are reduced.

[Tutorial Note: The International Telegraph and Telephone Consultative Committee uses the term 'quantising' in place of quantisation.]

Q6 For a CEPT 30-channel pulse-code modulation (PCM) system, (a) explain the purpose of

- a channel sampling gate,
- the encoder, and
- synchronisation insertion;

(b) sketch a typical 8 bit line signal at

- the output of the encoder, and
- the input to a pulse regenerator,

and hence state why it is necessary to use regeneration in a PCM system; and

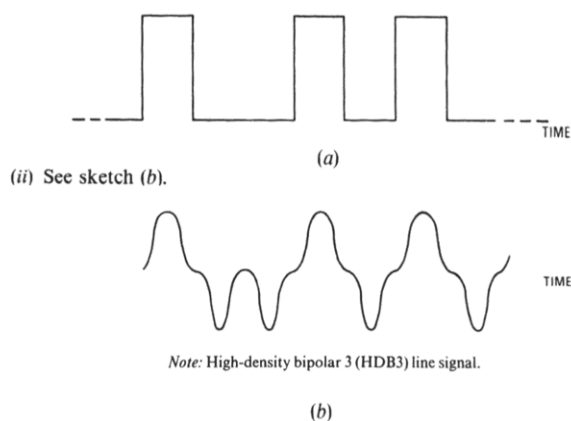
(c) state a typical distance between regenerators.

A6 (a) (i) The channel sampling gate performs two functions in a single operation. It samples each channel, producing a pulse-amplitude modulated (PAM) version of it, and then time-division multiplexes the sampled channels by gating them in turn onto a common highway, known as the *PAM highway*.

(ii) the encoder operates in turn on the samples appearing on the PAM highway and, for each one, generates an 8 bit code word; that is, a pulse-code modulated signal.

(iii) The synchronisation insertion enables each time-slot in the binary stream to be correctly identified. A *frame-alignment* word is inserted at time-slot 0, from which all the other time-slots can be found. When the *frame-alignment* signal is identified, the receiver is said to be in *synchronisation*.

(b) (i) See sketch (a).



Regenerators are necessary to prevent the cumulative build up of distortion and noise by restoring the signal in sketch (b) to its original undistorted state.

(c) 1.83 km.

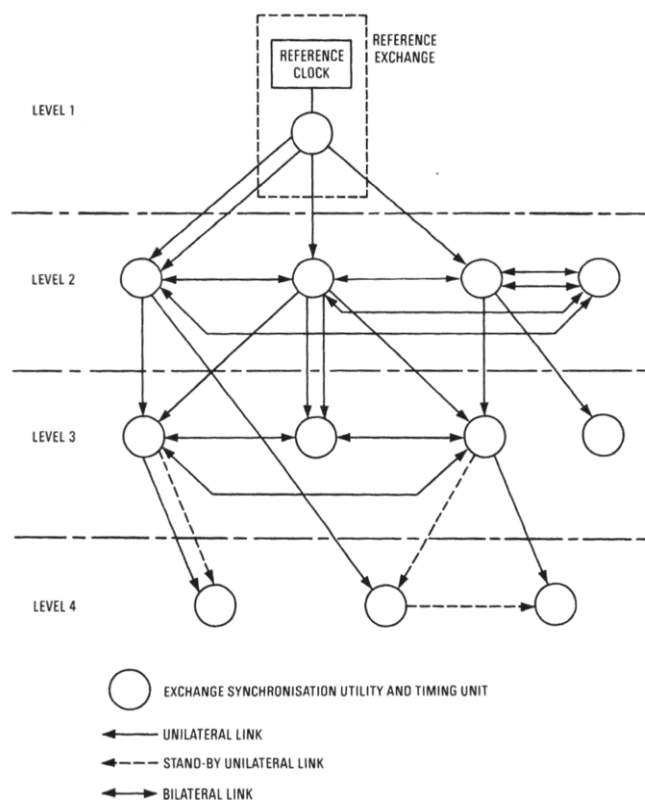
Q7 For a digital switching system

- define synchronous and asynchronous working,
- explain with the aid of a block diagram how synchronisation between switching nodes can be achieved, and
- discuss the advantages and disadvantages of synchronous working compared to asynchronous working.

A7 (a) In a synchronous digital switching system, the mean frequency difference between any two exchange timing units (TUs) is controlled so that it is zero. This is done by locking the TU to a reference clock signal distributed over a synchronisation network.

In an asynchronous digital switching system, each exchange has an independent TU which is periodically tuned to a reference clock to keep its drift within tolerable limits. The reference clock may be derived from a reference exchange in the highest level of the routing hierarchy, from a highly accurate and stable portable oscillator, preferably to caesium standard, or from a distributed reference signal similar to that used on the FDM network.

(b) Synchronisation between switching nodes in the UK is based on mutual synchronisation, where exchange clocks are interconnected by a synchronisation network so that each exchange is locked to the average



of all incoming clock rates. Each node in the network will be connected to other nodes by synchronisation links arranged in a four-tier hierarchy (see sketch). Synchronisation information is recovered from the frame-alignment patterns on nominated 2 Mbit/s systems. As can be seen from the diagram, control between exchanges in different levels is unilateral (that is, one way) from the higher to the lower. Within a level, control is bilateral. A synchronisation unit (SU) in the exchange compares the phase of the local timing unit (TU) with that of the nominated incoming digital streams. If a phase change is detected, the SU can send a control signal, for example, speed up or speed down to the local TU or to the distant exchange via digit 5 of time-slot 0 in the odd frames of the 2 Mbit/s digital path.

(c) Where there is a frequency difference between the incoming signal rate and the exchange TU, data is lost or repeated. This is known as *slip*, and its occurrence is a measure of the quality of service (QOS) offered. A synchronous system keeps slip occurrence low, whereas an asynchronous system would require frequent re-tuning to achieve the same QOS. Also, the setting accuracy becomes more critical for frequent re-tunes. However, if the QOS of an asynchronous system is acceptable, it is the cheaper alternative, initially at least. Nevertheless, cost advantages are derived from a synchronous system because it enables the design complexity of multiplexers to be reduced.

Q8 For a uniform transmission line:

- Define characteristic impedance (Z_0).
- State an expression for characteristic impedance in terms of its primary line coefficients.
- Draw a T-network diagram to represent the above line.
- A line has the following primary coefficients at an angular frequency of 5000 rad/s.

$$R = 30 \Omega/\text{km} \quad L = 5 \text{ mH}/\text{km}$$

$$G = 100 \mu\text{S}/\text{km} \quad C = 0.03 \mu\text{F}/\text{km}$$

Calculate the characteristic impedance for the line.

- Explain how characteristic impedance can be calculated from the results of two separate impedance measurements.

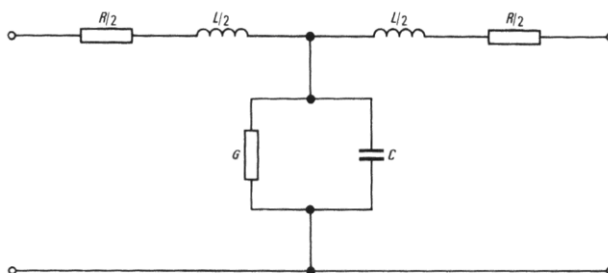
A8 (a) The characteristic impedance of a line is the impedance of an infinite length of the line, or the impedance of a line that is terminated with an impedance equal to the characteristic impedance.

(b) The characteristic impedance, Z_0 , is given by

$$Z_0 = \sqrt{\frac{R + j\omega L}{G + j\omega C}}$$

where R is the resistance,
 L is the inductance,
 G is the conductance,
 C is the capacitance, and
 ω is $2\pi \times$ frequency.

- See sketch



- Converting the expression for Z_0 in part (b) to polar form gives:

$$Z_0 = \sqrt{\frac{R^2 + \omega^2 L^2}{G^2 + \omega^2 C^2}} \angle \frac{1}{2} \left(\tan^{-1} \frac{\omega L}{R} - \tan^{-1} \frac{\omega C}{G} \right)$$

Substituting the given figures into this expression gives.

$$|Z_0| = \sqrt{\frac{900 + 25 \times 10^6 \times 25 \times 10^{-6}}{10^4 \times 10^{-12} + 25 \times 10^6 \times 9 \times 10^{-16}}}$$

$$= \sqrt{\frac{900 + 625}{1 \times 10^{-8} + 225 \times 10^{-10}}}$$

$$= \sqrt{\frac{1525}{3.25 \times 10^{-8}}}$$

$$= \sqrt{469.23 \times 10^8}$$

$$= 465.4 \Omega.$$

The phase angle is calculated as follows:

$$\begin{aligned} & \frac{1}{2} \left(\tan^{-1} \frac{\omega L}{R} - \tan^{-1} \frac{\omega C}{G} \right) \\ &= \frac{1}{2} \left\{ \tan^{-1} \left(\frac{5000 \times 5 \times 10^{-3}}{30} \right) - \tan^{-1} \left(\frac{5000 \times 0.03 \times 10^{-6}}{100 \times 10^{-6}} \right) \right\} \\ &= \frac{1}{2} (\tan^{-1} 0.833 - \tan^{-1} 1.5), \\ &= \frac{1}{2} (39.8^\circ - 56.3^\circ), \\ &= \frac{1}{2} (-16.5^\circ) \\ &= -8.25^\circ. \end{aligned}$$

Thus,

$$Z_0 = 465.4 \angle -8.25^\circ \Omega.$$

(e) Z_0 can be calculated from the expression

$$Z_0 = \sqrt{(Z_{oc} Z_{sc})}$$

where Z_{oc} is the measured impedance with the far end of the line open-circuited, and Z_{sc} is the measured impedance with the far end of the line short-circuited.

Q9 A uniform transmission line 3 km long is terminated in its characteristic impedance, $Z_0 = 1200 \angle -30^\circ \Omega$. At a certain frequency, the attenuation coefficient α of the line is 3 dB/km and its phase change coefficient β is 0.2 rad/km.

(a) If the sending end sinusoidal voltage is 1 V RMS, find

- the amplitude of the received current
- the phase of the received current relative to the sending-end voltage.

(b) Draw a phasor diagram representing the sending-end currents and voltages.

A9 (a) (i) If V_s is the sending-end voltage, then the sending-end current, I_s , is given by

$$I_s = \frac{V_s}{Z_0} = \frac{1 \times 10^3}{1200} = 0.83 \text{ mA}.$$

The received current, I_R , is given by

$$I_R = I_s \times 10^{-\alpha l/20},$$

where α is the attenuation coefficient in decibels/kilometre, and l is the line length.

Hence,

$$I_R = 0.83 \times 10^{-(3 \times 3)/20} = 0.83 \times 10^{-0.45} = 0.294 \text{ mA}.$$

(ii) By using the sending end voltage, V_s , as the phase reference, that is, $V_s = 1 \angle 0^\circ \text{ V}$, the sending current

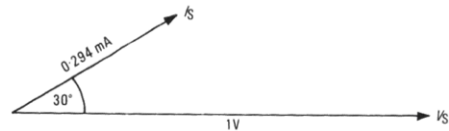
$$= \frac{V_s}{Z_0} = \frac{1 \times 10^3 \angle 0^\circ}{1200 \angle -30^\circ} = 0.83 \angle 30^\circ \text{ mA}.$$

Therefore, the sending current leads the sending voltage by 30° . As the phase-change coefficient is 0.2 rad/km, the received current lags the sending current by

$$0.2 \times 3 = 0.6 \text{ rad} = 34.37^\circ.$$

The received current therefore lags the sending voltage by

$$30^\circ - 34.37^\circ = -4.37^\circ.$$



Q10 (a) For the generation of light in an optical-fibre system explain the basic principles of

- a light-emitting diode (LED), and
- a laser diode.

(b) State THREE advantages of the laser diode over the LED.

(c) Explain why the LED is more suitable than the laser diode for use in graded-index multimode fibres.

(d) State typical output powers for laser and LED light sources.

(e) State typical bit-rate capacities for multimode and monomode fibres.

A10 (a) (i) An LED is a forward biased p-n junction semiconductor made from gallium arsenide (GaAs). This material has the property of emitting infra-red light when current passes through the junction.

(ii) A laser diode is a more complex junction made of the same basic material as an LED but with the ends polished and parallel. When the current through the junction exceeds a threshold level, light is reflected between the polished ends, and laser action occurs.

- Narrower spectrum,
- the beam diameter is more closely matched to fibre, and
- higher power.

(c) Graded-index fibres are designed to support a wider bandwidth than monomode fibres, but scattering limits the length of the fibre and so power is less of a consideration.

(d) Typical output powers for GaAs laser diodes are between 1–4 mW. For LEDs, they are between 50–150 μW .

(e) Multimode fibres are generally used for bit rates below 140 Mbit/s. Monomode fibres are commonly used for bit rates of 140 Mbit/s and above.

[Tutorial Note: Monomode fibre is also known as single-mode fibre.]

Answers contributed by B. McGrath and D. J. Parsons